## REMARKS

Reconsideration and allowance are respectfully requested.

Claims 1-9 and 17-34 stand rejected under 35 U.S.C. §102 as being anticipated by USP 5,970,053 to Schick. This rejection is respectfully traversed.

To establish that a claim is anticipated, the Examiner must point out where each and every limitation in the claim is found in a single prior art reference. *Scripps Clinic & Research Found. v. Genentec, Inc.*, 927 F.2d 1565 (Fed. Cir. 1991). Every limitation contained in the claims must be present in the reference, and if even one limitation is missing from the reference, then it does not anticipate the claim. *Kloster Speedsteel AB v. Crucible, Inc.*, 793 F.2d 1565 (Fed. Cir. 1986). Schick fails to satisfy this rigorous standard.

Schick relates to a digital channel cable TV transmission system. Schick is concerned with clipping distortion and describes controlling peak factor conditions within coherent FDM systems. The Examiner focuses on the signal analysis portion of Figure 1 in which a radio frequency (RF) signal detected at coupler 166 is converted to intermediate frequency (IF) at a mixer 168 using a local oscillator signal from a local oscillator 170. The IF signal is filtered in an anti-aliasing filter 182, and the filtered IF signal is synchronized with a sync signal (related to the sampling signal sent to the ADC 192) in a summer 190 before being converted to digital form in the ADC 192. The digital signal is then processed in the DSP 244.

Schick describes that the anti-aliasing filter 182 is necessary:

First, although  $S_{IF}$  (t) (output of mixer 168) represents a down-converted and band-limited subset of  $S_{RF}$  (t) (within composite signal 120), it still contains high frequency components which would alias when digitally sampled by the Analog-to-Digital Converter (ADC) 192. This simply means that frequency components higher than half the sampling frequency appear as low frequency artifacts in a Fast Fourier Transform (FFT) spectrum,

PETERSSON et al. Appl. No. 10/091,596 May 25, 2005

thereby interfering with genuine low frequency component measurements. The *IF BPF bank 182 thus primarily operates as an anti-aliasing filter*, but also serves to attenuate unwanted low frequency mixing products as well as video and audio modulation signals. (Emphasis added). Col. 8, lines 49-61.

Thus, Schick is just like the configurations described in the background of the instant application that the inventors recognized could be improved by eliminating the need for such anti-aliasing filters.

The background section explains that anti-aliasing filters suppress the spectral components that are outside the Nyquist band, i.e., out-of-band frequency components caused by the mixing and sampling, (i.e., the aliased signals), that would otherwise contaminate the analog-to-digital converter output. This aliasing effect may still occur even when such a filter is used, if it is not sufficiently accurately designed or manufactured. An example of this aliasing effect is illustrated using the frequency spectrums shown in Figs. 1A-1G.

Fig. 1F shows the spectral result of the output of the mixer when the fundamental and the odd harmonics of the local oscillator are mixed with the desired signal in the receive band. A filter characteristic, represented as the thick-line trapezoid, can remove those unwanted spectral components shown in dotted lines, leaving only the desired signal at a0, A0 along with those undesired spectral components that were not filtered, e.g., b0, B0. Unfortunately, these in-band, third or higher order harmonics cause undesired aliasing in the analog-to-digital conversion.

Fig. 1G shows a sampled mixer output from an analog-to-digital converter, when a non-optimal filter, such as that shown in Fig. 1F, is used to filter the mixer output. The undesired, aliased signals are shown as "dotted" spectrums, and the desired signals are shown as "hatched" spectrums. The sampling frequency of the analog-to-digital converter and its harmonics are shown as thick black vertical lines centered for each spectral copy and are indicated as  $F_{ADC}$ .

The aliasing problem is particularly troublesome when the analog-to-digital converter sampling rate is relatively high or higher than the local oscillator frequency. Although the sampling frequency satisfies the Nyquist sampling theorem and is more than twice the highest frequency of the receive band for the desired signal a0, A0, the sampling frequency is not more than twice the highest frequency of the mixer products b0, b0. As a result, those sampled, third harmonic signals b0, b0 are aliased into the frequency range of the desired signals. Being in the desired frequency band, the aliased signals cannot be readily removed by a filter, (e.g., a digital filter).

But anti-aliasing filters have disadvantages. First, they must be very accurately designed and constructed in order to eliminate the aliasing effect. To do this, the filter must be usually of a high order so that it has a sharp cutoff and a low ripple in the pass band in order to remove all signals except a0, A0 in Fig. 1F. Second, such filters are typically expensive, and even then, have certain variances and losses in the pass band that require compensation. Third, the filter must be matched to both source and load impedance in order to function properly. The mixer "source" impedance is typically low, and the analog-to-digital converter "load" impedance is typically high and slightly capacitive. Because high performance analog-to-digital converters usually suffer from decreasing linearity as the voltage swing over their input increases, it is beneficial to keep the load impedance as low as possible. Impedance matching is difficult because of the non-linear input impedance of the analog-to-digital converter, which means there is no fixed impedance value to use as a reference when calculating the filter impedance level. This non-linear impedance results in an anti-aliasing filter whose transfer function depends on the input signal's amplitude and frequency. As a result, the filter's bandwidth, insertion loss, and ripple will not match specifications for a fixed load impedance.

The present invention eliminates the need for an anti-aliasing filter while, at the same time, ensuring that aliasing in the pass band does not occur as a result of the mixing and sampling processes. The mixer's local oscillator frequency is related to the frequency of the analog-to-digital converter sampling rate in such a way so as to avoid aliasing in the desired receive band. In particular, the frequency of the local oscillator signal is an integer multiple of half of a sampling rate of the analog-to-digital converter. In a preferred, non-limiting, example embodiment, the sampling rate of the analog-to-digital converter and the frequency of the local oscillator are related by the following:  $F_{LO} = n * F_{ADC} / 2$ , where n is any positive integer.

Claim 1 now incorporates the feature of dependent claim 7 and recites:

wherein the apparatus is used in a receiver without a filter between the mixer and the analog-to-digital converter, and wherein a frequency of the local oscillator signal is set in relation to a frequency of a sampling rate of the analog-to-digital converter to avoid aliasing in a desired receive band.

Schick clearly shows an anti-aliasing filter 182 between mixer 168 and the ADC 192. As is made clear by Schick in the text quoted above, the anti-aliasing filter 182 is required; otherwise, undesired aliasing occurs. No where does Schick disclose setting the local oscillator frequency in relation to the frequency of a sampling rate of the analog-to-digital converter to avoid aliasing in a desired receive band. To avoid aliasing, Schick must use the anti-aliasing filter 182.

The Examiner makes reference to col. 6, lines 52-67 in Schick. These lines merely state the Nyquist theorem that "in order to preserve all frequency information when digitizing a continuous-time signal, the minimum sampling rate should be at least twice that of the highest frequency component of the signal being sampled." But this text relates to the sampling rate of the ADC 192. It does not describe relating the local oscillator frequency to the frequency of a sampling rate of the analog-to-digital converter to avoid aliasing in a desired receive band.

PETERSSON et al. Appl. No. 10/091,596 May 25, 2005

The Examiner also refers to col. 10, lines 1-26 in Schick. This text describes synchronizing the sampling operation. Schick explains at col. 9, lines 20-33:

The ADC 192 samples its input at a rate determined by signal  $S_{FS}$  (t) provided by signal generator 194 which is phase-locked to the 1.024 MHz system reference 108 applied thereto via signal path 110. Even so, data sampling can begin at any point relative to the IF signal components to be analyzed...It is therefore necessary to include some form of time synchronization signal,  $S_{SYNC}$  (t), such as that provided by signal generator 196, which is also phase-locked to the 1.024 MHz system reference 108 applied thereto via signal path 110.

Thus, the aliasing referred to in column 10 relates to the time synchronization signal,  $S_{SYNC}$  (t), which operates on the <u>already-filtered</u> IF signal at summer 190. The time synchronization signal,  $S_{SYNC}$  (t), is not used to prevent aliasing caused by the mixing operation at mixer 168.

Although not specifically addressed in the office action, claim 17 recites that "the analog, frequency-converted signal is connected directly to an input of the analog-to-digital converter."

Schick plainly shows that the IF output of the mixer 168 is connected to the ADC 192 by way of the intervening anti-aliasing filter 182, variable gain amplifier 262, and summer 190. Hence, Schick fails to disclose this feature of claim 17.

Claim 24 incorporates the feature of dependent claim 27 and now recites:

wherein the apparatus is used in a receiver without a filter between the mixer and the analog-to-digital converter, and wherein a frequency of the local oscillator signal is related to a sampling rate of the analog-to-digital converter to prevent aliasing that would otherwise result from the mixing and converting

Schick both requires a filter and lacks the recited relationship between the LO signal and the ADC sampling rate. Claim 30 recites similar features.

The anticipation rejection is improper and should be withdrawn. The application is now in condition for allowance. An early notice to that effect is earnestly solicited.

PETERSSON et al. Appl. No. 10/091,596 May 25, 2005

Respectfully submitted,

NIXON & VANDERHYE P.C.

By:

John R. Lastova Reg. No. 33,149

JRL:srd 901 North Glebe Road, 11th Floor Arlington, VA 22203-1808

Telephone: (703) 816-4000 Facsimile: (703) 816-4100